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Description

Method for switching voice traffic relations between a telephone communication network and an Internet

Internet terminals of an Internet, i.e. an Internet-capable communication network, frequently set up communication relations to the Internet via a telephone communication network, e.g. a public telephone network. The Internet terminal is implemented, for example, by means of a personal computer which is connected to the telephone communication network, e.g. a public telephone network, via a modem, the connection of the telephone communication network in most cases also being connected to a telephone terminal, for example a telephone. The Internet terminal sets up a communication relation or a connection, respectively, to an Internet server of an Internet provider directly with the aid of a modem equipped with a dialing method or with the aid of the telephone.

Voice information can be exchanged between the Internet terminals with the aid of the Voice over Internet Protocol known among experts and called VoIP in the further text. Such a VoIP is described, for example, in recommendations of various IETF workgroups (among others a recommendation for the Session Initiation Protocol SIP) or ITU recommendation H.323, the voice information being considerably compressed and inserted into Internet transmission packets. Furthermore voice connections can be set up between telephone terminals of a telephone communication network and Internet terminals with VoIP function. For this purpose, a gateway is provided with the aid of which the signaling of the telephone communication network is adapted to

the Internet signaling and the voice information contained in the transmission packets formed in

with the VoIP is converted into voice information according to the telephone communication network, and conversely. When a voice connection is set up from an Internet terminal to a telephone, its call numbers can be

specified directly by the Internet terminal and, in the case of a connection set-up the Internet terminal, a communication relation is first established with an SIP server in the case of IETF VoIP signaling or with a gatekeeper in the case of H.323 signaling. When it receives a call number which is allocated to a subscriber in the telephone communication network, the SIP server or gatekeeper sets up a communication relation to the gateway and from the latter a voice connection to the telephone communication network. In the case of a connection set-up from a telephone of the telephone communication network, signaling is conducted from the switching device connected to the gateway to the gateway. From the latter, a communication relation is set up to the SIP server or gatekeeper, with the aid of which the incoming call number is converted into an Internet-related terminal address. This terminal address is used for setting up a VoIP connection from the gateway to the Internet terminal determined by the terminal address.

In the case of a connection set-up from a telephone terminal to a further telephone terminal, it may happen that the called telephone terminal is busy, the busy state being caused by a connection or communication relation already existing to the Internet. In principle, however, a communication relation would be possible via the gateway and the Internet because a voice connection or voice traffic relation, respectively, is additionally possible during an Internet session with the Internet due to the VoIP function.

The object forming the basis of the invention consists in improving the possibility for voice traffic relations between the terminals of telephone communication networks and the Internet. The object is

achieved by the features of claim 1.

The essential aspect of the method according to the invention can be seen in the fact that a call diversion in the telephone communication network is set up by a telephone terminal before an

Internet session or by an associated Internet terminal during an Internet session, in such a manner that a connection set-up for a voice traffic relation initiated from a further telephone terminal to the telephone terminal is diverted to the associated Internet terminal. In this method, a uniform call number is advantageously used for terminals in the telephone communication network and in the Internet - claim 2. A significant advantage of the method according to the invention can be seen in the fact that no changes need to be carried out in the existing telephone communication network and the function for a call diversion in the telephone communication network, set up by the Internet terminal, is possible with minimum additional expenditure since signaling from the Internet to the telephone communication network is already implemented in a gateway or server and must only be expanded by setting a call diversion. A further advantage of the method according to the invention is the improved availability of terminals when setting up voice traffic relations or voice connections between the telephone communication network and the Internet.

According to a further embodiment of the method according to the invention, the call diversion is set up by an Internet terminal by signaling via a gateway to the telephone communication network, the signaling being converted in the gateway - claim 4. This signaling is already implemented for voice traffic relations or voice connections, respectively, between the Internet and the telephone communication network and only needs to be supplemented by the capability of setting the call diversion. This addition can be provided with very little additional expenditure.

As an alternative, the call diversion can be set up by an Internet terminal by signaling via a subscriber

server and an intelligent communication network connected to the latter and to the telephone communication network

- claim 5. According to a further alternative, the call diversion is set up by an Internet terminal by signaling via a subscriber server and a packet switching communication network connected to the latter and the telephone communication network - claim 6. In this alternative, the packet switching communication network is implemented, for example, in accordance with the X.25 standard and information is exchanged, for example, in accordance with a TCP/IP protocol. Signaling between the respective Internet terminal and the subscriber server takes place in accordance with Internet signaling and the signaling in the subscriber server is converted into the signaling in the intelligent communication network. The signaling of the intelligent communication network is adapted to the signaling in the telephone communication network - claim 7. The communication relation from the Internet terminal to the subscriber server is controlled via the web pages normally used in the Internet, i.e. a web-page-based interface to the Internet is implemented in the subscriber server. The connection between the subscriber server and the telephone communication network is advantageously established via an intelligent network, in which arrangement the signaling between the telephone communication network and the intelligent communication network is already implemented and the interface in the subscriber server can be implemented with little additional expenditure.

Further advantageous embodiments of the method according to the invention and a communication arrangement for carrying out the method according to the invention can be found in the further claims.

In the text which follows, the invention will be explained with reference to two drawings, in which:

Figure 1 shows a communication arrangement for implementing the invention in a block diagram

Figure 2 shows a communication relation between a telephone terminal and an Internet terminal in a flowchart.

Figure 1 shows an Internet INT which is implemented, for example, in accordance with the ITU standard H.323 "Packet-based multimedia communications systems". As an alternative, not shown, the Internet INT can be implemented in accordance with the IETF standard. A gatekeeper GK, with the aid of which the Internet addresses are converted and accesses between the Internet terminals IKE via the Internet INT and to a gateway GW connected to the Internet INT are controlled, is provided for call control in the Internet INT implemented in accordance with the H.323 standard.

Furthermore, a call number server CFS and an authentication server RAD is connected to the Internet INT. In the call number server CFS, the Internet addresses for the Internet terminals IKE are stored, i.e. even the addresses for diverted Internet terminals. Call control is carried out with the aid of "presence information" which is implemented by a "presence service". The presence information can be an Internet address or an E-mail address or a call number according to the telephone communication network, and if E-mail addresses are specified, the voice information is packetized into Internet packets and transmitted to the Internet terminal IKE having the specified E-mail address. Additionally, a dialog box for communicating with the call number server CFS, which is provided for dealing with voice traffic relations, is integrated in the Internet terminal. The dialog box makes it possible to hold an incoming voice connection SPV i.e. a voice over Internet connection - or to divert it to a voice mail box or to transmit a

busy signal to the calling terminal IKE.

The authentication server RAD implements a security function in the Internet INT. In this arrangement, information to be checked such as, for example, passwords, are transmitted from the respective Internet terminal IKE to the authentication server RAD where they are verified and the result of the check is transmitted to the Internet terminal IKE. The authentication server RAD can also be used by other servers for checking information, the authentication server RAD being implemented in accordance with the RFC 2138 and RFC 2139 standards in the case of an IETF Internet. In a further administration server ADS connected to the Internet INT, the administration functions such as, for example, configuration management, administration of access rights and determination of charges are implemented.

In the method according to the invention, in order to also provide for a voice traffic relation between a telephone terminal FE and an Internet terminal IKE connected to the Internet INT via a telephone communication network FEN and currently conducting an Internet session or subsequently conducting an Internet session, a call diversion CF is set in the telephone communication network FEN or, respectively, the associated communication system KS, for the Internet terminal IKE, i.e. for the associated connection of the telephone communication network FEN. This setting causes a connection set-up initiated by a telephone terminal FE, which is actually to be switched to the telephone terminal FE, the associated Internet terminal IKE of which is currently conducting an Internet session via the telephone connection, to be diverted to the Internet INT or, respectively, to the Internet terminal IKE conducting the Internet session.

For implementing the setting of a call diversion CF according to the invention, a subscriber server ISC is preferably provided which is connected both to the Internet INT and to an intelligent communication network IN or directly to a communication system KS of the telephone communication network YFEN.

The subscriber server ISC also communicates with a communication system KS of the telephone communication network FE via the intelligent communication network IN, communication with the communication systems KS of the telephone communication network FEN taking place, for example, via the standardized signaling network SS7 in the intelligent communication network IN. In the communication systems KS, a standardized interface to the intelligent communication network IN is in each case provided which is defined as service switching point SSP. A communication relation is established via this SSP interface to a standardized service control point SCP of the intelligent communication network IN, this service control point SCP being implemented in the subscriber server ISC. The information causing the call diversion CF to be set for the affected connection in the telephone communication network FEN is formed in the subscriber server ISC in accordance with the SS7 signaling and transmitted to the affected communication system KS via the intelligent communication network IN.

As an alternative, communication takes place not via an intelligent communication network IN but, for example, in accordance with a TCP/IP protocol (transmission control protocol/Internet protocol). The transport protocol is here implemented, for example, in accordance with the X.25 packet switching protocol, i.e. transmission between a communication system KS of the telephone communication network FEN and the subscriber server ISC takes place via an X.25 connection or, respectively, an X.25 communication network X.25, communication systems KS frequently having an X.25 communication connection for remote operation and remote maintenance. As an alternative, communication between the subscriber server ISC and telephone communication network FE can also take place

via the standardized Q3 interface. The information provided for the communication systems KS for setting the call diversion CF is adapted to the operational man/machine language MML of the communication systems KS

used in the telephone communication network FEN, e.g. the EWSD switching system by Siemens, i.e. the call diversion CS is set for the telephone terminals FE affected in the telephone communication network FE with the aid of the subscriber server ISC.

According to the invention, an Internet terminal IKE conducting a current Internet session sets a call diversion CF in the telephone communication network FE with the aid of the subscriber server ISC, the call diversion CF being set for the associated connection in the telephone communication network FE via which the current Internet session is being conducted. As an alternative, the call diversion CF can be set by the associated telephone terminal FE before the Internet session FE which is to be currently conducted. This assumes that the traffic relation from the associated telephone terminal FE of the telephone communication network FE to the Internet INT is established via an access device ISP of an Internet service provider. In this arrangement, the connection set-up is initiated and performed by the telephone terminal FE and, after a connection set-up via the telephone communication network FE and via the access device ISP to the Internet INT, a personal computer PC inserted between the telephone communication network FE and the telephone terminal FE initiates an Internet session. In this method, the subscriber line ASL is controlled by a modem function MOD implemented by a modem, with the aid of which the digital signals or data transmitted by the personal computer PC are converted into analog signals which can be transmitted via the telephone communication network FE and conversely. As an alternative, the access function and signaling of the telephone terminal FE can be integrated in a modem function MOD - i.e. in a modem - of a personal computer PC, the user interface of the personal computer PC also

being used for setting up and clearing down the connection.

Furthermore, a gateway GW, with the aid of which the Voice over Internet function VoIP of the Internet is converted into telephone communication network protocols, is provided for intercommunication between the Internet INT and the telephone communication network FE. Advantageously, the standardized signaling No. 7 - indicated by the designation SS7 in figure 1 - is used between the gateway GW and the telephone communication network FEN and the physical interface is implemented by a PCM interface PCM - indicated by the designation PCM in figure 1.

In the text which follows, the setting of a call diversion CF according to the invention from an Internet terminal IKE in a communication system KS or, respectively, in the telephone communication network FEN is described, it being assumed that a communication relation or an Internet session has been set up with the aid of the associated telephone terminal FE. To use the Voice over Internet function VoIP in the Internet INT, an Internet terminal IKE initiates registration at its gatekeeper GK - at the session initiation protocol (SIP) GK in an alternative solution - with the aid of the administration server ADS. In this process, it is indicated to the administration server ADS that the Internet terminal IKE is actively conducting an Internet session and the Voice over Internet function VoIP is possible on its personal computer PC. The registration is carried out by log-in in the Internet INT, the administration server ADS checking whether the Internet terminal IKE is allowed to access the Voice over Internet function VoIP. For this purpose, the administration server ADS calls up the authentication server RAD in order to verify the authentication for the Voice over Internet function VoIP. The authentication server RAD transmits a positive or negative confirmation to the administration

server ADS in accordance with the result of the check. Following this, the Internet terminal IKE performs a registration at the gatekeeper GK. After this registration, the administration server ADS provides an Internet address for the relevant

Internet terminal IKE and stores it in the presence service which forwards this Internet address to the gatekeeper GK and to the call number server CFS.

Following this, a web page is opened in the Internet terminal IKE by means of which a communication relation to the subscriber server ISC is initiated. After checking its authentication, the Internet terminal IKE affected can change its call diversion information CFA in the database. The call diversion information CFA is transmitted via the intelligent communication network IN to the associated communication system KS of the telephone communication network FN where it is stored in its database with the aid of a call diversion routine CFR - indicated by a rectangle marked by CFR in figure 1, as a result of which a call diversion CF is set for the telephone connection or, respectively, the telephone terminal FE, the associated Internet terminal IKE of which is currently conducting an Internet session.

In the text which follows, a connection set-up for a voice traffic relation - called voice connection SPV in the further text - from a telephone terminal FE to an Internet terminal IKE is described with the aid of a flowchart in figure 2, the called Internet terminal IKE conducting an Internet session and a call diversion CF being set for its connection in the telephone communication network FEN in accordance with the preceding registration.

In figure 2, a dashed vertical line is in each case specified for the telephone terminal FE affected, the communication network FEN or communication system KS, respectively, the gateway GW, the gatekeeper GK and the Internet terminal IKE affected or the personal computer

PC respectively, the dashed lines indicating the progress in time of the signaling described in figure 2.

The calling telephone terminal FE transmits a call information item CALL to the telephone communication network FEN and the communication system KS affected diverts this call information item CALL to the gateway GW due to the call number rn specified in this information item. This call diversion CF is carried out on the basis of the call diversion information cfa stored in the communication system KS for the connection, determined by the call number rn, of the telephone communication network FEN or the telephone terminal FE connected to it. The gateway GW sets up a communication relation KB to the gatekeeper GK of the Internet INT. In the gatekeeper GK, a check is made whether the Internet terminal IKE determined by the call number rn is allowed to use the Voice over Internet function VoIP. If there is no authorization for the called Internet terminal IKE - indicated by N in figure 2 -, the gatekeeper GK transmits a busy information item BUSY via the gateway GW and the telephone communication network FEN to the telephone terminal FE affected - indicated by arrows designated by BUSY in figure 2. If there is an authorization for the Voice over Internet function VoIP - indicated by a Y in figure 2 - , a check is then made in the gatekeeper GK whether the called Internet terminal IKE is currently conducting an Internet session. If the called Internet terminal IKE is not conducting an Internet session, a busy information item BUSY is transmitted via the gateway GW and the telephone communication network FEN to the telephone terminal FE - indicated by arrows designated by BUSY in figure 2. If the called Internet terminal IKE is in an Internet session - indicated by a Y in figure 2 - a Voice over Internet connection is set up with the aid of the gatekeeper GK to the Internet terminal IKE affected - indicated by an arrow designated by CONNECT in figure 2. This connection set-up sets up a voice connection

SPV between the telephone terminal FE and the Internet terminal IKE determined by the call number rn, the voice connection SPV being implemented by a dedicated voice connection between the gateway GW

and the telephone terminal FE and as a Voice over Internet connection between the gateway GW and the Internet terminal IKE.

Using the method according to the invention, a diversion of a connection set-up initiated from a telephone terminal FE to an Internet terminal IKE, which is in an Internet session, can thus be performed without changes in the telephone communication network FEN, the Internet terminal IKE being connected to the Internet INT via a telephone connection, i.e. like a telephone terminal FE via the telephone communication network FEN. A telephone terminal FE also associated with this telephone connection is disconnected from the telephone connection during the Internet session and is inactive, i.e. cannot be reached.

The method according to the invention is not restricted to the exemplary embodiment and can also be used in other implementations of the Internet INT and of the telephone communication network FEN - for example the ISDN communication network -, in which case the servers, the gateway, the access device and the intelligent communication network are to be adapted to the physical and procedural characteristics of the respective communication networks.